

ABSTRACT

The present invention relates to an apparatus and method for maintaining voice call quality over a packet network by providing optimal de-jitter buffer depth and rate of change of depth. Buffer depth and rate of change of buffer depth may be initially determined by classifying the incoming call. Classification of the incoming calls may be accomplished by categorizing calls into groups based on characteristics of the calls. The buffer depth and rate of change of depth may be further optimized at the start of calls based on voice-path delay and packet loss probability measurements over one or more calls of the same class such that the voice-path delay is minimized while maintaining a certain packet loss probability, the packet loss probability is minimized while maintaining a certain voice-path delay, or an R-factor, which is an objective measure of voice quality, is maximized.